



Application Notes for Configuring Avaya Communication Server 1000E R7.5 with Avaya Aura[®] Session Manager R6.1 and Avaya Session Border Controller for Enterprise R4.0.5 to support BT Global Services NOAS SIP Trunk - Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between an Avaya SIP enabled enterprise solution and the BT Global Services NOAS SIP Trunk service. The Avaya solution consists of Avaya Aura[®] Session Manager and Avaya Communication Server 1000E connected to an Avaya Session Border Controller for Enterprise. BT is a member of the Global SIP Service Provider program.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the necessary steps to configure Session Initiation Protocol (SIP) trunking between an Avaya SIP enabled enterprise solution and the BT SIP Trunk Service. The Avaya solution consists of Avaya Aura[®] Session Manager, an Avaya Communication Server 1000E (CS1000E) and an Avaya Session Border Controller for Enterprise (Avaya SBCE) connected to the BT SIP Trunk Service. Customers using this Avaya SIP enabled enterprise solution with the BT SIP Trunk Service are able to place and receive PSTN calls via a dedicated Internet connection using the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. The approach normally results in lower cost and a more flexible implementation for the enterprise customers.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager, Communication Server 1000E and the Avaya SBCE. The enterprise site was configured to use the SIP Trunk Service provided by BT, with all PSTN traffic transiting via the BT SIP Trunk Service.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by BT. Incoming PSTN calls were terminated on Digital, Unistim, SIP and Analogue telephones at the enterprise side.
- Outgoing calls from the enterprise site were completed via BT to PSTN telephones. Outgoing calls from the enterprise to the PSTN were made from Digital, Unistim, SIP and Analogue telephones.
- Calls were made using G.729A, and G.711A codecs.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 transmission mode.
- DTMF transmission using RFC 2833 with successful IVR menu progression.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by BT requiring Avaya response and sent by Avaya requiring BT response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the BT SIP Trunk Service with the following observations:

- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Routing to emergency numbers (such as 112) was tested.
- G.729 annex b (silence suppression) is not supported by BT SIP Trunk Service and thus was not tested.
- Unsupervised transfer of incoming or outgoing PSTN calls to PSTN called parties is not permitted; this is a PSTN imposed restriction. The same restriction exists for supervised transfers of an existing PSTN call to a PSTN called party.
- Mobile-X call to service DN, then making a call out to the PSTN. The call rings for a second and is dropped immediately, a BYE is sent from NOAS.

2.3. Support

For technical support on BT products please use the following web link.

<http://btbusiness.custhelp.com/app/contact>

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to BT's SIP Trunk Service. Located at the Enterprise site is a Session Border Controller, Session Manager and CS1000E. Endpoints are Avaya 1140 series IP telephones, Avaya 1200 series (not shown in **Figure 1**) IP telephones (with Unistim and SIP firmware), Avaya IP Softphones (SMC3456, 2050 and Avaya one-X® Communicator), Avaya Digital telephone, Analogue telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

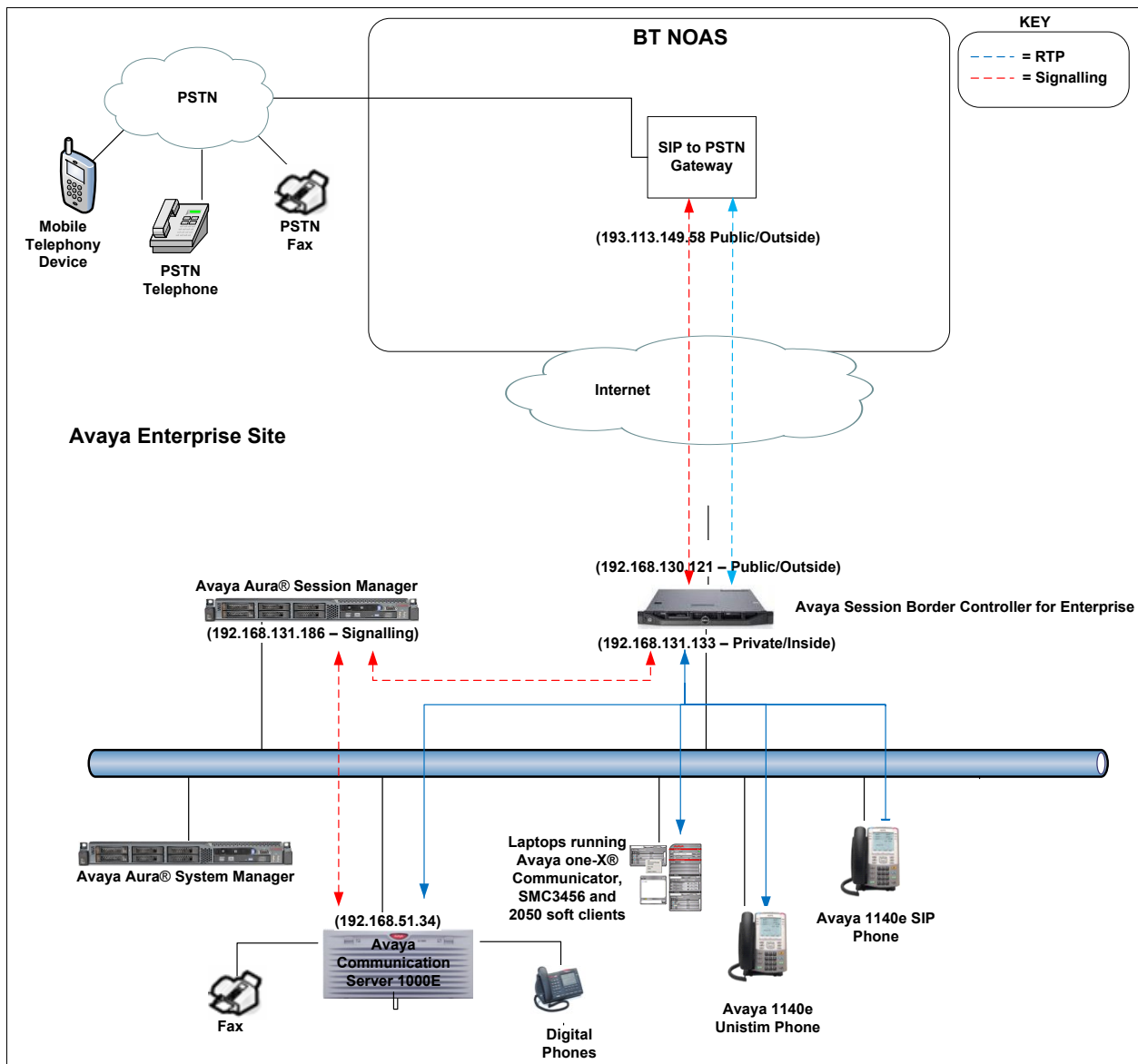


Figure 1: Test Setup BT SIP Trunk Service to Avaya Enterprise

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya Communication Server 1000E	Avaya Communication Server 1000E 007.50Q/7.50.17 (PSWV 100 with latest Patches and Deplist as shown in Appendix A)
Avaya Communication Server 1000E Media Gateway	CSP Version: MGCC CD01 MSP Version: MGCM AB01 APP Version: MGCA BA07 FPGA Version: MGCF AA18 BOOT Version: MGCB BA07 DSP1 Version: DSP1 AB04
Avaya S8800 Server running Avaya Aura® Session Manager	Avaya Aura® Session Manager 6.1 (6.1.3.0.613006)
Avaya S8800 Server running Avaya Aura® System Manager	Avaya Aura® System Manager 6.1 (6.1.7.1.1260)
Dell R210 V2 Server running Avaya Session Border Controller for Enterprise	Avaya Session Border Controller for Enterprise (4.0.5.Q02)
Avaya 1140e Unistim Phone	FW: 0625C8J
Avaya 1140e SIP Phone	FW: 4.00.04.00
Avaya one-X® Communicator	Version cs6.1.0.25
Avaya SMC3456	Version 2.6 Build 57666
Avaya Analogue Telephone	N/A
Avaya M3904 Digital Telephone	N/A
BT NOAS SIP Trunking	3.120.5.17

5. Configure Avaya Communication Server 1000E

This section describes the steps required to configure Communication Server 1000E for SIP Trunking and also the basic configuration for telephones (Analogue, SIP and IP phones). SIP trunks are established between Communication Server 1000E and Session Manager. SIP trunks are also established between Session Manager and the Avaya SBCE private interface. The Avaya SBCE public interface connects to the BT Global Services NOAS SIP trunks. Incoming PSTN calls from the BT Global Services NOAS SIP Trunk service traverse the Avaya SBCE and are directed to the Session Manager, which directs the calls to Communication Server 1000E (see **Figure 1**).

The Avaya SBCE media manager has been configured to ensure RTP packets are managed correctly from the Avaya SBCE public interface to the private interface and vice versa. When a SIP message arrives at Communication Server 1000E, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Server 1000E and may be first subject to outbound features such as route selection, digit manipulation and class of service restrictions. When Communication Server 1000E selects a SIP trunk for outgoing PSTN calls, SIP signaling is directed to the Session Manager. The Session Manager directs the outbound SIP messages to the Avaya SBCE private interface. The Avaya SBCE public interface manages outgoing SIP sessions onwards to the BT Global Services NOAS SIP trunks.

Specific Communication Server 1000E configuration was performed using Element Manager and the system terminal interface. The general installation of the Avaya Communication Server 1000E, System Manager, Session Manager and Avaya SBCE is presumed to have been previously completed and is not discussed here. Configuration details will be provided as required to draw attention to changes in default system configurations.

5.1. Logging into the Avaya Communication Server 1000E

Log in using SSH to the ELAN IP address of the call server using a user with correct privileges. Once logged in, type **csconsole**, this will take the user into the VxWorks shell of the call server. Next type **logi**, the user will then be asked to login with correct credentials. Once logged in, the user can then progress to load any overlay.

5.2. Confirm System Features

The keycode installed on the call server controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the Communication Server 1000E system terminal and manually load overlay 22 to print the System Limits (the required command is SLT), and verify that the number of SIP Access Ports reported by the system is sufficient for the combination of trunks to the BT network, and any other SIP trunks needed. See the following screenshot for a typical System Limits printout. The value of **SIP ACCESS PORTS** defines the maximum number of SIP trunks for the Communication Server 1000E.

```
System type is - Communication Server 1000E/CP PM
CP PM - Pentium M 1.4 GHz
```

IPMGs Registered:			4		
IPMGs Unregistered:			0		
IPMGs Configured/unregistered:			2		
TRADITIONAL TELEPHONES	120	LEFT	110	USED	10
DECT USERS	16	LEFT	16	USED	0
IP USERS	10000	LEFT	9954	USED	46
BASIC IP USERS	16	LEFT	13	USED	3
TEMPORARY IP USERS	8	LEFT	8	USED	0
DECT VISITOR USER	16	LEFT	16	USED	0
ACD AGENTS	192	LEFT	185	USED	7
MOBILE EXTENSIONS	8	LEFT	7	USED	1
TELEPHONY SERVICES	16	LEFT	13	USED	3
CONVERGED MOBILE USERS	8	LEFT	8	USED	0
AVAYA SIP LINES	16	LEFT	12	USED	4
THIRD PARTY SIP LINES	16	LEFT	16	USED	0
PCA	20	LEFT	18	USED	2
ITG ISDN TRUNKS	0	LEFT	0	USED	0
H.323 ACCESS PORTS	524	LEFT	524	USED	0
AST	6652	LEFT	6640	USED	12
SIP CONVERGED DESKTOPS	16	LEFT	16	USED	0
SIP CTI TR87	16	LEFT	8	USED	8
SIP ACCESS PORTS	524	LEFT	518	USED	6
RAN CON	90	LEFT	90	USED	0
MUS CON	120	LEFT	120	USED	0

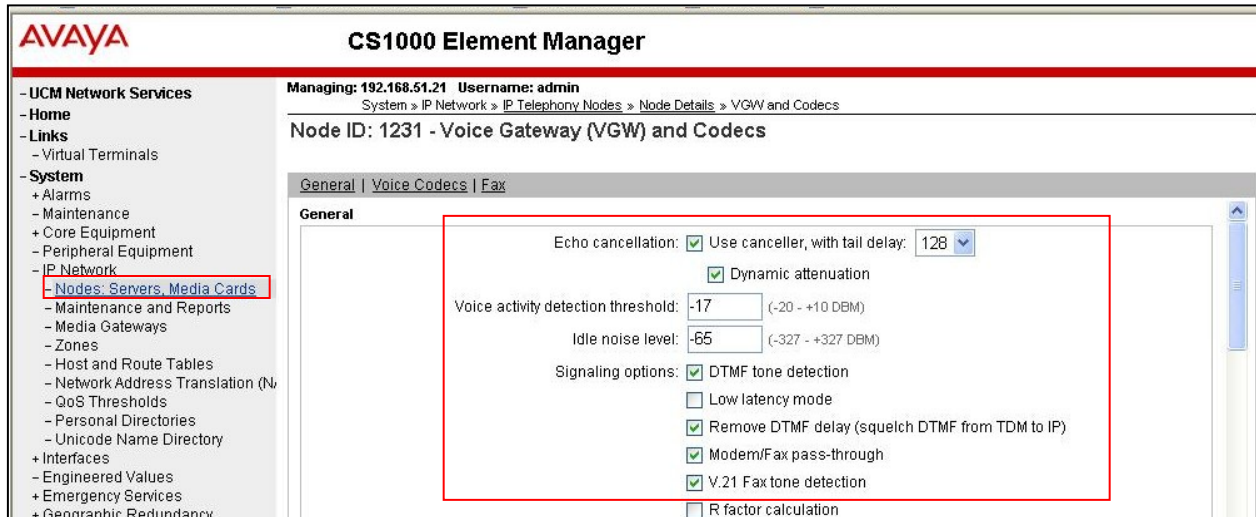
Load overlay 21, and confirm the customer is setup to use ISDN trunks (see below).

```
REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

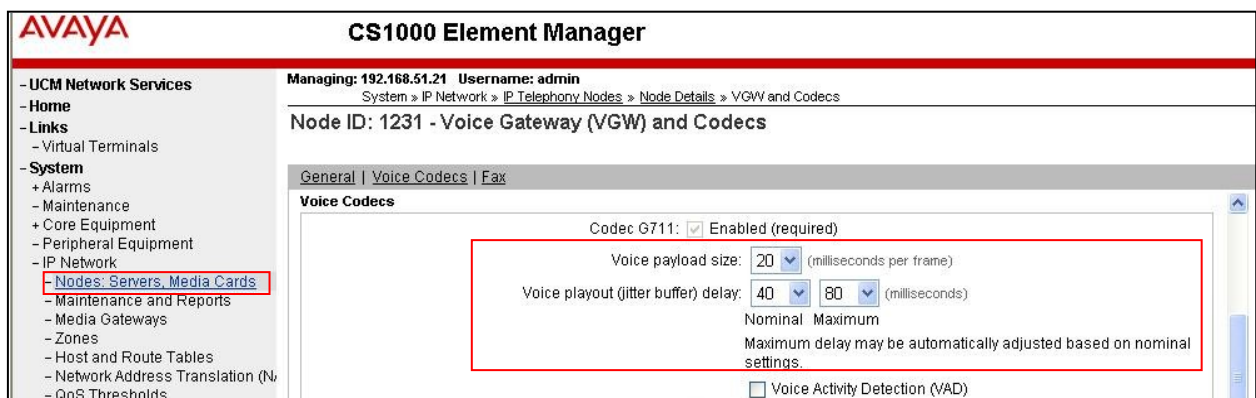
TYPE NET_DATA
CUST 00
OPT RTD
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
ISDN YES
```

5.3. Configure Codecs for Voice and FAX operation

The BT Global Services NOAS SIP Trunk service supports G.711A and G.729A voice codecs and T.38 FAX transmissions. Using the Communication Server 1000E element manager sidebar, select **Nodes, Servers, Media Cards**. Navigate to the **IP Network → IP Telephony Nodes → Node Details → VGW and Codecs** property page and configure the Communication Server 1000E General codec settings as in the following screenshots. The values highlighted are required for correct operation. The following screenshot shows the necessary **General** settings.



Move down to the Voice Codecs section and configure the G.711 codec settings. The following screenshot shows the G.711 codec settings.



Next, scroll down to the G.729 codec section and configure the settings.

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'Nodes: Servers, Media Cards' highlighted. The main content area is titled 'Node ID: 1231 - Voice Gateway (VGW) and Codecs'. Under the 'Voice Codecs' tab, the 'Codec G729' section is highlighted with a red box. It shows 'Codec G729: Enabled'. Below this, 'Voice payload size' is set to 30 (milliseconds per frame) and 'Voice playback (jitter buffer) delay' is set to 60 (Nominal) and 120 (Maximum) milliseconds. A note states: 'Maximum delay may be automatically adjusted based on nominal settings.' At the bottom of this section, there is a checkbox for 'Voice Activity Detection (VAD)' which is currently unchecked.

Finally, configure the Fax settings as in the highlighted section of the next screenshot. Click on the **Save** button when finished.

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'Nodes: Servers, Media Cards' highlighted. The main content area is titled 'Node ID: 1231 - Voice Gateway (VGW) and Codecs'. Under the 'Fax' tab, the 'Codec name: T.38 FAX' section is highlighted with a red box. It shows 'Maximum rate' set to 14400 (bps), 'Fax TCF method' set to 2, 'Fax playback nominal delay' set to 100 (0 - 300 milliseconds), and 'FAX no activity timeout' set to 20 (10 - 32000 milliseconds). Below this, 'Packet size' is set to 30 (bps). A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' There are 'Save' and 'Cancel' buttons at the bottom right.

5.4. Virtual Trunk Gateway Configuration

Use Communication Server 1000E Element Manager to configure the system node properties. Navigate to the **System → IP Networks → IP Telephony Nodes → Node Details** and verify the highlighted section is completed with the correct IP addresses and subnet masks.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 1231 - SIP Line, LTPS, PD, IP Media Services, Gateway (SIPGw, H323Gw))

Node ID: * (0-9999)

Call server IP address: * TLAN address type: IPv4 only
 IPv4 and IPv6

Embedded LAN (ELAN) **Telephony LAN (TLAN)**

Gateway IP address: * Node IPv4 address: *

Subnet mask: * Subnet mask: *

Node IPv6 address:

* Required Value.

Associated Signaling Servers & Cards

Select to add [Print](#) | [Refresh](#)

<input type="checkbox"/>	Hostname ▲	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/>	primflwr-leads	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	192.168.51.19	192.168.51.36	Follower
<input type="checkbox"/>	primleader-leads	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	192.168.51.18	192.168.51.35	Leader

Show: IPv6 address

The next screenshot shows the SIP Virtual Trunk Gateway configuration, navigate to **System → IP Networks → IP Telephony Nodes → Node Details → Virtual Trunk Configuration Details** and in the **General** area, fill in the highlighted areas with the relevant settings.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System > IP Network > IP Telephony Nodes > Node Details > Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Vtrk gateway application: Enable gateway service on this node

General

Vtrk gateway application: SIPGw and H.323Gw
SIP domain name: umlab.local
Local SIP port: 5060 *(1 - 65535)
Gateway endpoint name: PRIM_SS_LEADER
Gateway password: *
H.323 ID: PRIM_SS_LEADER
Application node ID: 1231 *(0-9999)
Enable failsafe NRS:

Virtual Trunk Network Health Monitor

Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below.

Monitor IP: Add

Monitor addresses:
192.168.131.186
192.168.51.46
Remove

Scroll down to the **Proxy or Redirect Server** area and fill in the values for **Proxy Server Route 1**. The Primary **TLAN IP address**, **Port** and **Transport protocol** values are required.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System > IP Network > IP Telephony Nodes > Node Details > Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 192.168.131.186
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

Options: Support registration
 Primary CDS proxy

Secondary TLAN IP address: 0.0.0.0
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

Move down the page and fill in **Tertiary IP address**, **Port** and **Transport protocol** (see the next screenshot). Fill in the **Proxy Server Route 2** values including **Primary TLAN IP address**, **Port** and **Transport protocol**.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Tertiary IP address: 192.168.51.169
Port: 5060 (1 - 65535)
Transport protocol: TCP

Options: Support registration
 Tertiary CDS proxy

Proxy Server Route 2:

Primary TLAN IP address: 192.168.131.186
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)
Transport protocol: TCP

Options: Registration not supported
 Primary CDS proxy

Scroll down to the CLID Presentation section and fill in the **Country code (CCC)** and **Area code** values as shown below.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Options: Registration not supported
 Primary CDS proxy

CLID Presentation:

Country code (CCC): 44
Area code: 113 NPA in North America

Move to the **SIP URI Maps** section and fill in the values (see next screenshot).

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 1231 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

SIP URI Map:

Public E.164 domain names		Private domain names	
National:	E164.Nat	UDP:	udp
Subscriber:	E164.Sub	CDP:	cdp.udp
Special number:	PublicSpecial	Special number:	PrivateSpecial
Unknown:	PublicUnknown	Vacant number:	PrivateUnknown
		Unknown:	UnknownUnknown

Scroll down to the bottom of the page and click on the **Save** button (not shown), then save and transmit (not shown).

5.5. Configure Bandwidth Zones

Bandwidth Zones are used for alternate call routing between IP stations and for Bandwidth Management. SIP trunks require a unique zone that are not shared with other resources and best practice dictates that IP telephones and Media Gateways are all placed in a separate zone than SIP trunks. Use Element Manager to define bandwidth zones as in the following highlighted example. Use Element Manager and navigate to **Zones** → **System** → **IP Network** → **Zones** → **Bandwidth Zones** and add new zones as required.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System » IP Network » Zones » Bandwidth Zones

Bandwidth Zones

Add... Edit... Import... Export Maintenance... Delete Refresh

Zone #	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1	100000	BQ	100000	BB	SHARED	MO	GR_PRIM
2	100000	BQ	100000	BB	SHARED	MO	GR_SEC
3	100000	BQ	10000	BB	SHARED	MO	SURV_MG1000
4	1000000	BQ	1000000	BQ	SHARED	VTRK	SIPLINEZONE
5	1000000	BQ	1000000	BB	SHARED	VTRK	SIP_VTRK_NOAS
6	100000	BQ	10000	BQ	SHARED	MO	VIRTUALSETS
7	100000	BQ	100000	BQ	SHARED	VTRK	VIRTUAL_TRKS

5.6. Configure SIP Trunks

Communication Server 1000E virtual trunks will be used for all inbound and outbound PSTN calls to the BT SIP Trunk Service. Five separate steps are required to configure Communication Server 1000E virtual trunks:-

- Configure a D-Channel Handler (**DCH**); configure using the Communication Server 1000E system terminal and overlay 17.
- Configure a SIP trunk Route Data Block (**RDB**); configure using the Communication Server 1000E system terminal and overlay 16.
- Configure SIP trunk members; configure using the Communication Server 1000E system terminal and overlay 14.
- Configure a Route List Block (**RLB**); configure using the Communication Server 1000E system terminal and overlay 86.
- Configure Special Prefix Numbers (**SPN's**); configure using the Communication Server 1000E system terminal and overlay 90.

The following is an example DCH configuration for SIP trunks. Load overlay 17 at the Communication Server 1000E system terminal and enter the following values. The highlighted entries are required for correct SIP trunk operation. Exit overlay 17 when completed.

```
Overlay 17
ADAN      DCH 50
CTYP DCIP
DES  VIR TRK
USR  ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA YES
IFC SL1
CNEG 1
RLS  ID 5
RCAP ND2
MBGA NO
H323
    OVLR NO
    OVLS NO
```

Next, configure the SIP trunk Route Data Block (RDB) using the Communication Server 1000E system terminal and overlay 16. Load overlay 16, enter **RDB** at the prompt, press return and commence configuration. The value for **DCH** is the same as previously entered in overlay 17. The value for **NODE** should match the node value in **Section 5.3**. The value for **ZONE** should match that used in **Section 5.4** for **SIP_VTRK_NOAS**. The remaining highlighted values are important for correct SIP trunk operation.

<pre> Overlay 16 TYPE: rdb CUST 00 ROUT 100 TYPE RDB CUST 00 ROUT 100 DES VIR_TRK TKTP TIE NPID_TBL_NUM 0 ESN NO RPA NO CNVT NO SAT NO RCLS EXT VTRK YES ZONE 00253 PCID SIP CRID NO NODE 1231 DTRK NO ISDN YES MODE ISLD DCH 50 IFC SL1 PNI 00001 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC NO ISAR NO DAPC NO MBXR NO MBXOT NPA MBXT 0 PTYP ATT CNDP UKWN AUTO NO DNIS NO DCDR NO ICOG IAO SRCH LIN TRMB YES STEP </pre>	<pre> ACOD 1600 TCPP NO PII NO AUXP NO TARG CLEN 1 BILN NO OABS INST IDC YES DCNO 10 NDNO 10 * DEXT NO DNAM NO SIGO STD STYP SDAT MFC NO ICIS YES OGIS YES TIMR ICF 1920 OGF 1920 EOD 13952 LCT 256 DSI 34944 NRD 10112 DDL 70 ODT 4096 RGV 640 GTO 896 GTI 896 SFB 3 PRPS 800 NBS 2048 NBL 4096 IENB 5 TFD 0 VSS 0 VGD 6 EESD 1024 SST 5 0 DTD NO SCDT NO 2 DT NO NEDC ORG FEDC ORG </pre>	<pre> CPDC NO DLTN NO HOLD 02 02 40 SEIZ 02 02 SVFL 02 02 DRNG NO CDR NO NATL YES SSL CFWR NO IDOP NO VRAT NO MUS YES MRT 21 PANS YES RACD NO MANO NO FRL 0 0 FRL 1 0 FRL 2 0 FRL 3 0 FRL 4 0 FRL 5 0 FRL 6 0 FRL 7 0 OHQ NO OHQT 00 CBQ NO AUTH NO TTBL 0 ATAN NO OHTD NO PLEV 2 OPR NO ALRM NO ART 0 PECL NO DCTI 0 TIDY 1600 100 ATTR NO TRRL NO SGRP 0 ARDN NO CTBL 0 AACR NO </pre>
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Next, configure virtual trunk members using the Communication Server 1000E system terminal and overlay 14. Configure sufficient trunk members to carry both incoming and outgoing PSTN calls. The following example shows a single SIP trunk member configuration. Load overlay 14 at the system terminal and type **new X**, where X is the required number of trunks. Continue entering data until the overlay exits. The **RTMB** value is a combination of the **ROUT** value entered in the previous step and the first trunk member (usually 1). The remaining highlighted values are important for correct SIP trunk operation.

```

Overlay 14
TN 160 0 0 0
DATE
PAGE
DES VIR_TRK
TN 160 0 00 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 00253
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK ANLG
NCOS 0
RTMB 100 1
CHID 1
TGAR 1
STRI/STRO WNK WNK
SUPN YES
AST NO
IAPG 0
CLS TLD DTN CND ECD WTA LPR APN THFD XREP SPCD MSBT
P10 NTC
TKID
AACR NO

```

Configure a Route List Block (RLB) in overlay 86. Load overlay 86 at the system terminal and type **new**. The following example shows the values used. The value for **ROUT** is the same as previously entered in overlay 16. The **RLI** value is unique to each RLB.

<pre> Overlay 86 CUST 0 FEAT rlb RLI 24 ELC NO ENTR 0 LTER NO ROUT 100 TOD 0 ON 1 ON 2 ON 3 ON 4 ON 5 ON 6 ON 7 ON VNS NO SCNV NO CNV NO EXP NO FRL 0 DMI 0 </pre>		<pre> CTBL 0 ISDM 0 FCI 0 FSNI 0 BNE NO DORG NO SBOC NRR PROU 1 IDBB DBD IOHQ NO OHQ NO CBQ NO ISET 0 NALT 5 MFRL 0 OVLL 0 </pre>
---	--	---

Next, configure Special Prefix Number(s) (SPN) which users will dial to reach PSTN numbers. Use the Communication Server 1000E system terminal and overlay 90. The following are some example SPN entries used. The highlighted **RLI** value previously configured in overlay 86 is used as the Route List Index (RLI). This is the default PSTN route to the SIP Trunk service.

SPN 999	SPN 90	SPN 2	SPN 15
FLEN 3	FLEN 7	FLEN 7	FLEN 3
ITOH NO	ITOH NO	ITOH NO	ITOH NO
CLTP NONE	CLTP NONE	CLTP NONE	CLTP NONE
RLI 24	RLI 24	RLI 24	RLI 24
SDRR NONE	SDRR NONE	SDRR NONE	SDRR NONE
ITEI NONE	ITEI NONE	ITEI NONE	ITEI NONE

5.7. Configure Analogue, Digital and IP Telephones

A variety of telephone types were used during the testing. The following is the configuration for the Avaya 1140e Unistim IP telephone. Load overlay 20 at the system terminal and enter the following values. A unique five digit number is entered for the **KEY 00** and **KEY 01** value. The value for **CFG_ZONE** is the value used in **Section 5.4** for **VIRTUALSETS**.

```

Overlay 20 IP Telephone configuration
DES 1140
TN 096 0 01 16 VIRTUAL
TYPE 1140
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00254
CUR_ZONE 00254
ERL 0
ECL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
LNRS 16
XLST
SCPW
SFLT NO
CAC MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMD LLCN MCTD CLBD AUTR
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTA AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
FDSO NOVQ VOLA VOUD CDMR PRED RECA MCDD T87D SBMD KEM3 MSNV FRA PKCH MUTA MWTD

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```

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```
DVLD CROD CROD
CPND_LANG ENG
RCO 0
HUNT 0
LHK 0
PLEV 02
PUID
DANI NO
AST 00
IAPG 1
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 52000 0      MARP
      CPND
        CPND_LANG ROMAN
          NAME IP1140
          XPLN 10
          DISPLAY_FMT FIRST, LAST
01 MCR 52000 0
      CPND
        CPND_LANG ROMAN
          NAME IP1140
          XPLN 10
          DISPLAY_FMT FIRST, LAST

02
03 BSY
04 DSP
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
```

Digital telephones are configured using the overlay 20; the following is a sample 3904 digital set configuration. Again, a unique number is entered for the **KEY 00** and **KEY 01** value.

Overlay 20 - Digital Set configuration

```
TYPE: 3904
DES 3904
TN 000 0 09 08 VIRTUAL
TYPE 3904
CDEN 8D
CTYP XDLC
CUST 0
MRT
ERL 0
FDN 0
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 1
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR PUA MTD FND HTD TDD HFA GRLD CRPA STSD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMA LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXA ARHD FITD CNTD CLTD ASCD
CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
FSDS NOVD CDMR PRED RECA MCDD T87D SBMD PKCH CROD CROD
CPND LANG ENG
RCO 0
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 1
AACS
ACQ
ASID
SFNB
SFRB
USFB
CALB
FCTB
ITNA NO
DGRP
PRI 01
MLWU_LANG 0
```

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```
MLNG ENG
DNDR 0
KEY 00 MCR 52001 0      MARP
      CPND
        CPND_LANG ROMAN
          NAME Digital Set
          XPLN 10
          DISPLAY_FMT FIRST, LAST
01 MCR 52001 0
      CPND
        CPND_LANG ROMAN
          NAME Digital Set
          XPLN 10
          DISPLAY_FMT FIRST, LAST
02 DSP
03 MSB
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27 CLT
28 RLT
29
30
31
```

Analogue telephones are also configured using overlay 20; the following example shows an Analogue port configured for Plain Ordinary Telephone Service (POTS) and also configured to allow T.38 Fax transmission. A unique value is entered for **DN** which is the extension number. **DTN** is required if the telephone uses DTMF dialing. Values **FAXA** and **MPTD** configure the port for T.38 Fax transmissions.

```

Overlay 20 - Analogue Telephone Configuration
DES 500
TN 100 0 00 03
TYPE 500
CDEN 4D
CUST 0
MRT

ERL 00000
WRLS NO
DN 52002
AST NO
IAPG 0
HUNT
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
XLST
SCI 0
SCPW
SFLT NO
CAC_MFC 0
CLS UNR DTN FBD XFD WTA THFD FND HTD ONS
LPR XRD AGRD CWD SWD MWD RMMD SMWD LPD XHD SLKD CCSD LND TVD
CFTD SFD MRD C6D CNID CLBD AUTU
ICDD CDMD LLCN EHTD MCTD
GPUD DPUD CFXD ARHD OVDD AGTD CLTD LDTD ASCD SDND
MBXD CPFA CPTA UDI RCC HBTD IRGD DDGA NAMA MIND
NRWD NRCN NROD SPKD CRD PRSD MCRD
EXR0 SHL SMSD ABDD CFHD DNDY DNO3
CWND USMD USRD CCBN BNRD OCBN RTDD RBDD RBHD FAXA CNUD CNAD PGND FTTC
FDSD NOVD CDMR PRED MCDD T87D SBMD PKCH MPTD

PLEV 02
PUID
AACS NO
MLWU_LANG 0
FTR DCFW 4

```

5.8. Configure the SIP Line Gateway Service

SIP terminal operation requires the Communication Server node to be configured as a SIP Line Gateway (SLG) before SIP telephones can be configured. Prior to configuring the SIP Line node properties, the SIP Line service must be enabled in the customer data block. Use the Communication Server 1000E system terminal and overlay 15 to activate SIP Line services, as in the following example where **SIPL_ON** is set to **YES**.

```
SLS_DATA
SIPL_ON YES
UAPR 78
NMME NO
```

The numerical value entered for the **UAPR** setting will be pre-appended to all SIP Line phones, and is used internally to track SIP phones. Use Element Manager and navigate to the **IP Network → IP Telephony Nodes → Node Details → SIP Line Gateway Configuration** page. In the **General** section, configure the **SIP domain name**, **SLG Local Sip port** and **SLG Local Tls port**.

The screenshot displays the AVAYA CS1000 Element Manager interface. The main title is "AVAYA CS1000 Element Manager". The breadcrumb navigation is "System > IP Network > IP Telephony Nodes > Node Details > SIP Line Configuration". The page title is "Node ID: 1231 - SIP Line Configuration Details".

The interface is divided into several sections:

- General** (highlighted with a red box):
 - SIP domain name: *
 - SLG endpoint name:
 - SLG Group ID:
 - SLG Local Sip port: (1 - 65535)
 - SLG Local Tls port: (1 - 65535)
- Virtual Trunk Network Health Monitor**:
 - Monitor IP addresses (listed below)
 - Information will be captured for the IP addresses listed below.
 - Monitor IP: Add
 - Monitor addresses:
 -
 -Remove
- SIP Line Gateway Settings**:
 - Security policy:
 - Number of byte re-negotiation:
 - Options: Client authentication

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Scroll down to the **Branch /GR Office Settings** area. The IP address in **MO SLG IPv4 address** is the system **NODE IP address**, previously configured in **Section 5.3**. The **MO SLG port** and **MO SLG transport** values will be **5070** and **TCP**. Click on the **Save** button when finished.

AVAYA CS1000 Element Manager

Managing: 192.168.51.21 Username: admin
System > IP Network > IP Telephony Nodes > Node Details > SIP Line Configuration

Node ID: 1231 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Settings

Security policy: Best Effort
Number of byte re-negotiation: 0
Options: Client authentication
 x509 Certificate authentication enabled

SIP Line Gateway Service

Branch / GR Office Settings:

SLG role: MO
SLG mode: S1/S2
MO SLG IPv4 address: 192.168.51.34
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
MO SLG IPv6 address:
MO SLG port: 5070 (1 - 65535)
MO SLG transport: TCP

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

5.9. Configure SIP Line Telephones

When SIP Line service configuration is completed, use the Communication Server 1000E system terminal and overlay 20 to add a Universal Extension (UEXT). See the following example of a SIP Line extension. The value for **UXTY** must be **SIPL**. This example is for an Avaya SIP telephone, so the value for **SIPN** is 1. The **SIPU** value is the username, **SCPW** is the logon password and these values are required to register the SIP telephone to the SLG. The value for **CFG_ZONE** is the value set for **SIPLINEZONE** in **Section 5.4**. A unique telephone number is entered for value **KEY 00**. The value for **KEY 01** is comprised of the **UAPR** value (set to 78 previously in this section) and the telephone number used in **KEY 00**.

```
Overlay 20 - SIP Telephone Configuration
DES SIPD
TN 096 0 01 15 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TLSV 0
SIPU 52003
NDID 5
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID
NUID
NHTN
CFG_ZONE 00004
CUR_ZONE 00004
ERL 0
ECL 0
VSIT NO
FDN
TGAR 0
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 52003
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR MTD FNA HTA TDD HFD CRPD
MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXA ARHD FITD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
```


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UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD MSNV FRA PKCH MWTD DVLD
CROD CROD
CPND_LANG ENG
RCO 0
HUNT
LHK 0
PLEV 02
PUID
DANI NO
AST
IAPG 0 *

AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 MCR 52003 0 MARP
CPND
CPND_LANG ROMAN
NAME Sigma 1140
XPLN 11
DISPLAY_FMT FIRST, LAST*
01 HOT U 7852003 MARP 0
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23 *
24 PRS
25 CHG
26 CPN
27
28
29
30
31

6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Avaya Communication Server 1000E as a Managed Element

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the **Home** tab will be presented with menu options shown below.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 6.1", and links for "Help | About | Change Password | Log off admin". Below the title bar, there are two tabs: "Session Manager *" and "Home". The main content area is divided into three columns: "Users", "Elements", and "Services". The "Elements" column is highlighted with a red box around the "Routing" option, which includes "Network Routing Policy". Other options in "Elements" include "Application Management", "Communication Manager", "Conferencing", "Inventory", "Messaging", "Presence", "SIP AS 8.1", and "Session Manager". The "Users" column includes "Administrators", "Groups & Roles", "Subscribers", "Synchronize and Import", "UCM Roles", and "User Management". The "Services" column includes "Backup and Restore", "Configurations", "Events", "Licenses", "Replication", "Scheduler", "Security", "Templates", and "UCM Services".

6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the Elements Home tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button (not shown) to create a new SIP domain entry. In the **Name** field, enter the domain name (e.g., **umlab.local**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes.

Avaya Aura™ System Manager 6.1

Routing / Home

Home / Elements / Routing / Domains - Domain Management

Domain Management

Name	Type	Default	Notes
umlab.local	SIP	<input type="checkbox"/>	Avaya Blue CSTabs SIP Domain

Commit Cancel

6.3. Administer Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for the enterprise SIP entities. On the **Routing** tab, select **Locations** from the left hand menu. Under **General**, in the **Name** field enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**. Enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.

Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

General

* Name: Romford Avaya Lab

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Mbit/sec

Total Bandwidth: 2000

Per-Call Bandwidth Parameters

* Default Audio Bandwidth: 1 Mbit/sec

Location Pattern

Add Remove

IP Address Pattern	Notes
* [192.168.131.*]	

6.4. Administer Adaptations

Please note that the testing was performed in BT's lab in the UK. This lab is also being used for other work other than Compliance testing with the NOAS platform. There were two adaptations being used, one on the CS1000E and one on the Avaya SBCE. For completeness both adaptations will be described, however there is an overlap in what each adaptation is trying to achieve regarding digit manipulation.

6.4.1. Administer Adaptation for Communication Server 1000E

To ensure that the E.164 numbering format is used between the enterprise and BT SIP Trunk Service, an adaptation module is used to perform some digit manipulation. This adaptation is applied to the Communication Server 1000E SIP entity. To add an adaptation, under the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown).

Under **Adaption Details** → **General**:

- In the **Adaptation name** field, enter an informative name.
- In the **Module name** field, click on the down arrow and then select the **<click to add module>** entry from the drop down list and type **CS1000Adapter** in the resulting **New Module Name** field.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 6.1', and links for 'Help | About | Change Password | Log off admin'. The breadcrumb trail is 'Home / Elements / Routing / Adaptations - Adaptation Details'. The left sidebar shows a tree view with 'Routing' expanded, and 'Adaptations' highlighted with a red box. The main content area is titled 'Adaptation Details' and shows the 'General' tab. The 'Adaptation name' field contains 'adapt_PRIM_SS_LEADER' and is highlighted with a red box. The 'Module name' dropdown menu is open, showing 'CS1000Adapter' selected, and is also highlighted with a red box. Below these fields are input boxes for 'Module parameter:', 'Egress URI Parameters:', and 'Notes:'. At the top right of the form area are 'Commit' and 'Cancel' buttons, and a 'Help ?' link.

Scroll down the page and under **Digit Conversion for Incoming Calls to SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the **Matching Pattern** field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers received from Communication Server 1000E are converted to the E.164 numbering format before being processed by Session Manager. The following screenshot shows the settings used.

Digit Conversion for Incoming Calls to SM										
Add		Remove								Filter: Enable
12 Items Refresh										
<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes		
<input type="checkbox"/>	*003	*3	*36	PrivateSpecia	*2	+	destination	Ireland IDD Code		
<input type="checkbox"/>	*0113	*4	*36	PrivateSpecia	*1	+44	destination	Leeds Area STD Code		
<input type="checkbox"/>	*0121	*4	*36	PrivateSpecia	*1	+44	destination	Birmingham Area STD Code		
<input type="checkbox"/>	*0131	*4	*36	PrivateSpecia	*1	+44	destination	Edinburgh Area STD Code		
<input type="checkbox"/>	*01903	*5	*36	PrivateSpecia	*1	+44	destination	Worthing Area STD Code		
<input type="checkbox"/>	*0191	*4	*36	PrivateSpecia	*1	+44	destination	Tyneside Area STD Code		
<input type="checkbox"/>	*020	*3	*36	PrivateSpecia	*1	+44	destination	London Area STD Code		
<input type="checkbox"/>	*05	*2	*36		*0	+	both	Type:E164 Local, special rule		
<input type="checkbox"/>	*07	*2	*36	PrivateSpecia	*1	+44	destination	UK Mobile Services		
<input type="checkbox"/>	*x	*1	*36	cdp.udp	*0	55	both	Type:Level 0 Regional, special rule		
<input type="checkbox"/>	*x	*1	*36	PrivateSpecia	*0	56	both	Type:Special, general rule		
<input type="checkbox"/>	*x	*1	*36	+1	*0	+1	both	Type:E164 National, special rule		

Under **Digit Conversion for Outgoing Calls from Session Manager** click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the **Matching Pattern** field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers will have the + symbol and international dialing code removed before being presented to Communication Server 1000E. See the following screenshot for the settings used.

	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*#	* 1	* 36	udp	* 0		both	Type:Level 1 Regional Entity:PRIM
<input type="checkbox"/>	*+4420	* 5	* 36		* 3	0	destination	IC BT NOAS Call translation
<input type="checkbox"/>	*55	* 2	* 36	cdp.udp	* 2		both	Type:Level 0 Regional Entity:PRIM

6.4.2. Administer Adaptation for Avaya Session Border Controller for Enterprise

To ensure that the E.164 numbering format is used between the enterprise and BT SIP Trunk Service, an adaptation module is used to perform some digit manipulation. This adaptation is applied to the Avaya SBCE SIP entity. To add an adaptation, under the **Routing** tab select **Adaptations** on the left hand menu and then click on the **New** button (not shown). Under **Adaptation Details** → **General**:

- In the **Adaptation name** field enter an informative name.
- In the **Module name** field click on the down arrow and then select the **<click to add module>** entry from the drop down list and type **DigitConversionAdapter** in the resulting **New Module Name** field.

Home / Elements / Routing / Adaptations - Adaptation Details

Adaptation Details

General

* Adaptation name: Romford CM6.1 SIP stations

Module name: DigitConversionAdapter

Module parameter:

Egress URI Parameters:

Notes: to allow ddi calls to Rom 39xx SIP

Scroll down the page and under **Digit Conversion for Incoming Calls to SM**, click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the **Matching Pattern** field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers received by Session Manager are converted from E.164 numbering format before being processed and sent to Communication Server 1000E and terminated on to endpoints. The following screenshot shows the settings used.

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*+442079603250	* 13	* 36		* 13	53005	destination	
<input type="checkbox"/>	*+442079603251	* 13	* 36		* 13	252100	destination	
<input type="checkbox"/>	*+442079603252	* 13	* 36		* 13	252161	destination	

Under **Digit Conversion for Outgoing Calls from Session Manager** click the **Add** button and specify the digit manipulation to be performed as follows:

- Enter the leading digits that will be matched in the **Matching Pattern** field.
- In the **Min** and **Max** fields set the minimum and maximum digits allowed in the digit string to be matched.
- In the **Delete Digits** field enter the number of leading digits to be removed.
- In the **Insert Digits** field specify the digits to be prefixed to the digit string.
- In the **Address to modify** field specify the digits to manipulate by the adaptation. In this configuration the dialed number is the target so destination has been selected.

This will ensure any destination numbers will have the + symbol and international dialing code added before being presented to the Avaya SBCE. See the following screenshot for the settings used.

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
<input type="checkbox"/>	*+0	* 2	* 36		* 2	+44	destination	
<input type="checkbox"/>	*+00	* 3	* 36		* 3	+	destination	to allow international
<input type="checkbox"/>	*+1	* 3	* 36		* 1		destination	To allow DQ calls
<input type="checkbox"/>	*+44	* 3	* 36		* 1		origination	remove + for outbound to Hipcon
<input type="checkbox"/>	*+9	* 2	* 36		* 1		destination	To allow emergency calls
<input type="checkbox"/>	*90	* 2	* 36		* 2	+44	both	to allow sip phones to dial out

6.5. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu (see the following screenshot) and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity.

Under **SIP Entity Details** → **General**:

- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **Other** for a Communication Server 1000E SIP entity.
- In the **Location** field select the appropriate location from the drop down menu.
- In the **Time Zone** field enter the time zone for the SIP Entity.

In this enterprise site configuration there are three SIP Entities configured.

- Avaya Aura[®] Session Manager SIP Entity
- Communication Server 1000E SIP Entity
- Avaya Session Border Controller For Enterprise SIP Entity

6.5.1. Avaya Aura[®] Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signaling interface.

The screenshot shows the 'SIP Entity Details' configuration page. On the left is a navigation menu with options: Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and has a 'Commit' button in the top right. Under the 'General' tab, the following fields are visible: 'Name' (Romford SM 6.1), 'FQDN or IP Address' (192.168.131.186), 'Type' (Session Manager), 'Notes' (empty), 'Location' (Romford Avaya Lab), 'Outbound Proxy' (empty), 'Time Zone' (Europe/London), 'Credential name' (empty), and 'SIP Link Monitoring' (Use Session Manager Configuration). Red boxes highlight the Name, FQDN or IP Address, and Location fields.

The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop down menu select **rom2.bt.com** as the default domain.

The screenshot shows a configuration interface for ports. At the top, there are 'Add' and 'Remove' buttons. Below is a table with 3 items and a 'Refresh' button. The table has columns for 'Port', 'Protocol', 'Default Domain', and 'Notes'. Three rows are listed, each with a checkbox, a port number, a protocol dropdown, a default domain dropdown, and a notes field. The first row is (5060, TCP, rom2.bt.com), the second is (5060, UDP, rom2.bt.com), and the third is (5061, TLS, rom2.bt.com). A red box highlights the first two rows. Below the table, there is a 'Select : All, None' option and a '* Input Required' warning. A 'Commit' button is at the bottom right.

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	rom2.bt.com	
<input type="checkbox"/>	5060	UDP	rom2.bt.com	
<input type="checkbox"/>	5061	TLS	rom2.bt.com	

6.5.2. Avaya Communication Server 1000E SIP Entity

The following screenshot shows the SIP entity for Communication Server 1000E which is configured as **Type Other**. The **FQDN or IP Address** field is set to the Communication Server 1000E node IP address. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4.1**.

The screenshot shows the 'SIP Entity Details' configuration page. The left sidebar has a menu with 'SIP Entities' selected. The main area is titled 'SIP Entity Details' and 'General'. There are 'Commit' and 'Cancel' buttons at the top right. The configuration fields are:

- * Name: PRIM_SS_LEADER
- * FQDN or IP Address: 192.168.51.34
- Type: Other
- Notes: GR PRIME SITE
- Adaptation: adapt_PRIM_SS_LEADER
- Location: (empty dropdown)
- Time Zone: Europe/London
- Override Port & Transport with DNS SRV:
- * SIP Timer B/F (in seconds): 4
- Credential name: (empty text field)
- Call Detail Recording: none

 A red box highlights the 'Name', 'FQDN or IP Address', 'Type', and 'Notes' fields. Another red box highlights the 'Adaptation', 'Location', 'Time Zone', 'SIP Timer B/F', 'Credential name', and 'Call Detail Recording' fields.

6.5.3. Avaya Session Border Controller for Enterprise SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE private network interface configured in **Section 7** of this document. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4.2**.

Home / Elements / Routing / SIP Entities- SIP Entity Details

SIP Entity Details

General

* Name: 2nd_Romford_AASBC6.0

* FQDN or IP Address: 192.168.131.133

Type: Other

Notes: SIPERA SBC Romford being used f

Adaptation: Romford CM6.1 SIP stations

Location: Romford Avaya Lab

Time Zone: Europe/London

Override Port & Transport with DNS SRV:

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Commit Cancel Help ?

6.6. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select Session Manager.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.5**.
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.

Click **Commit** to save changes. The following screen shows the Avaya SBCE Entity Link used in this configuration.

Domains	Entity Links								Commit
Locations									
Adaptations									
SIP Entities									
Entity Links									
Time Ranges	1 Item Refresh								Filter:
Routing Policies	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes	
Dial Patterns	* Romford SM 6.1 to l	* Romford SM 6.1	UDP	* 5060	* Romford AASBC 6.0	* 5060	Trusted	link from Ses	
Regular Expressions									

The following screen shows the CS1000E Entity Link used in this configuration.

Domains	Entity Links								Help ?
Locations									Commit Cancel
Adaptations									
SIP Entities									
Entity Links									
Time Ranges	1 Item Refresh								Filter: Enable
Routing Policies	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes	
Dial Patterns	* PRIM_SS_LEADER t	* Romford SM 6.1	TCP	* 5060	* PRIM_SS_LEADER	* 5060	Trusted		
Regular Expressions									
Defaults									

6.7. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu (see next screenshot) and then click on the **New** button (not shown).

- Under **General** enter an informative name in the Name field.
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies.
- Under **Time of Day**, click **Add**, and then select the time range.

The following screen shows the routing policy for Communication Server 1000E. The **SIP Entity as Destination** value is set to **PRIM_SS_LEADER**, as entered in **Section 6.5.2**. The **Time of Day** is set to 24 hour by 7 day operation, this is not shown.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left navigation pane has 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and shows the 'General' tab. The 'Name' field is set to 'PRIM_SS_LEADER_Rank_1'. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with the following data:

Name	FQDN or IP Address	Type	Notes
PRIM_SS_LEADER	192.168.51.34	Other	GR PRIME SITE

The following screen shows the routing policy for the Avaya SBCE. The **SIP Entity as Destination** value is set to **2nd_Romford_AAASBCAE6.0**, as entered in **Section 6.5.3**. The **Time of Day** is set to 24 hour by 7 day operation, this is not shown.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left navigation pane has 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and shows the 'General' tab. The 'Name' field is set to 'Outbound calls to AASBC for NOA'. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with the following data:

Name	FQDN or IP Address	Type	Notes
2nd_Romford_AAASBC6.0	192.168.131.133	Other	SIPERA SBC Romford being used for NOAS

6.8. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu (see below) and then click on the **New** button (not shown).

Under **Dial Pattern Details** → **General**:

- In the **Pattern** field enter a dialed number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialed number.
- In the **Max** field enter the maximum length of the dialed number.
- In the **SIP Domain** field select the domain configured in **Section 6.2** or set to All.

Under **Originating Locations and Routing Policies**, click **Add**, in the resulting screen (not shown) under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.7**. Click **Select** button to save. The following screen shows an example dial pattern configured for the CS1000E. This dial pattern will route the calls to the CS1000E endpoints.

The screenshot displays the 'Dial Pattern Details' configuration page. On the left is a navigation menu with 'Dial Patterns' highlighted. The main area is titled 'Dial Pattern Details' and has 'General' selected. A red box highlights the 'General' configuration fields: Pattern (252), Min (3), Max (36), Emergency Call (unchecked), SIP Domain (-ALL-), and Notes (CS1000 Extn Range). Below this is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with one entry. The table has columns for Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy Disabled, Routing Policy Destination, and Routing Policy Notes. The entry shows '-ALL-' for the location name, 'Any Locations' for notes, 'PRIM_SS_LEADER_Rank_1' for the policy name, '1' for rank, 'Routing Policy Disabled' (unchecked), and 'PRIM_SS_LEADER' for the destination.

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-	Any Locations	PRIM_SS_LEADER_Rank_1	1	<input type="checkbox"/>	PRIM_SS_LEADER	

The following screen shows an example dial pattern configured for the Session Border Controller which will route the calls out to BT's SIP Trunk Service NOAS.

The screenshot displays the 'Dial Pattern Details' configuration page. The 'General' tab is active, showing the following fields:

- * Pattern:** +0
- * Min:** 2
- * Max:** 36
- Emergency Call:**
- SIP Domain:** -ALL-
- Notes:** Outgoing calls from enterprise to NOAS via 9

Below the general settings, there is a section for 'Originating Locations and Routing Policies' with an 'Add' button and a 'Remove' button. A table lists one policy:

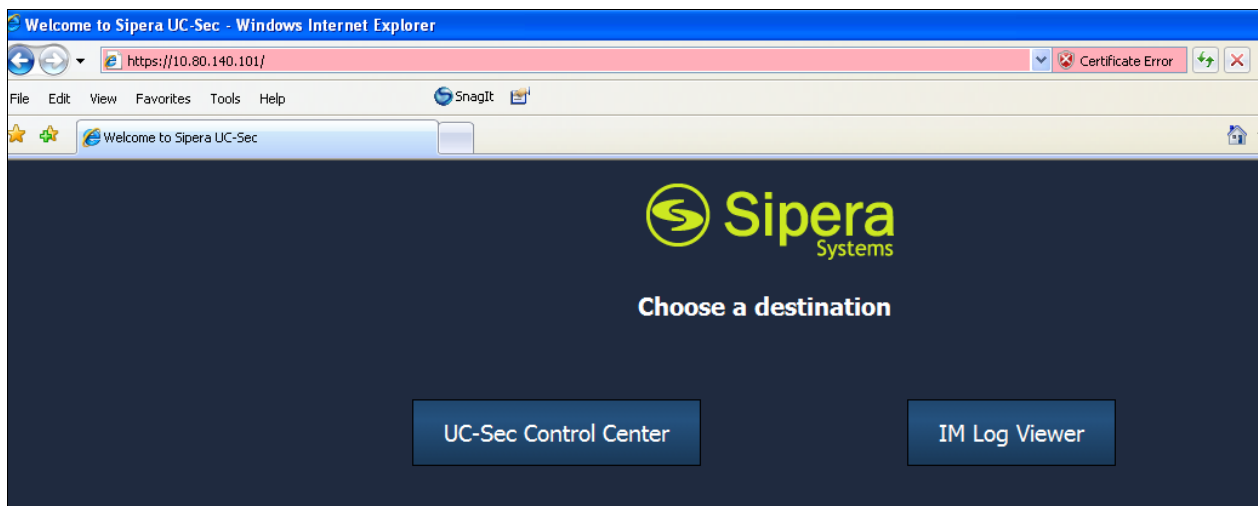
Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-	Any Locations	Outbound calls to AASBC for NOAS	5	<input type="checkbox"/>	2nd_Romford_AASBC6.0	

7. Avaya Session Border Controller for Enterprise Configuration

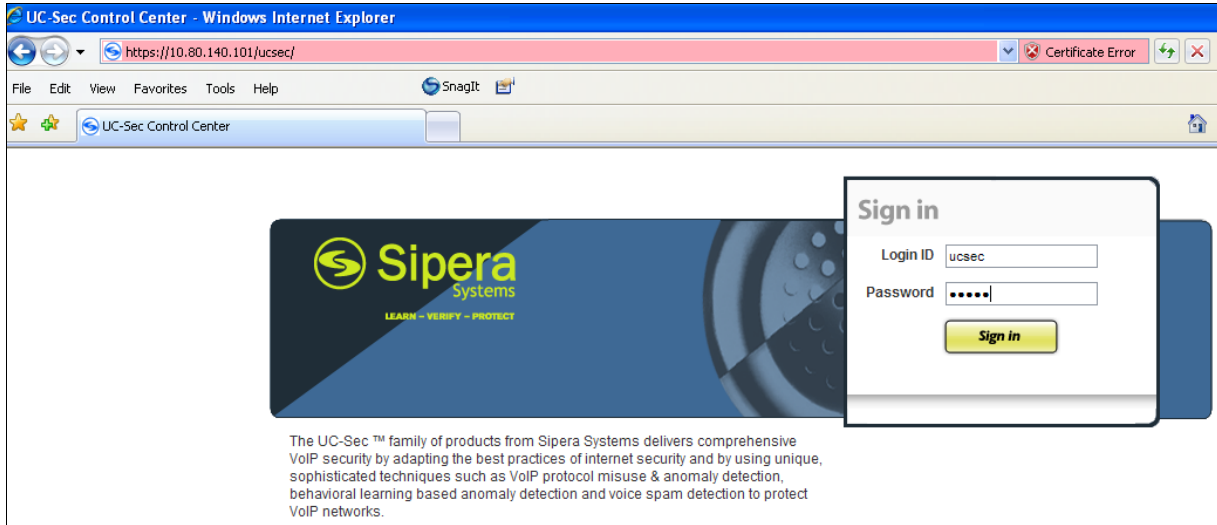
This section provides the procedures for configuring Session Border Controller for Enterprise.

7.1. Accessing UC-Sec Control Centre

Access the web interface by typing <https://x.x.x.x> (where x.x.x.x is the management IP of the Avaya SBCE).



Select **UC-Sec Control Center** and enter the **Login ID** and **Password**.



7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

7.2.1. Server Internetworking Avaya Side

Server Internetworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T.38. From the left hand menu select **Global Profiles** → **Server Interworking** and click on **Add Profile**.

- Enter **Profile Name: ToASM** and click **Next**.
- Check **Hold Support= RFC2543**.
- Check **T.38 Support**.
- All other options on the General Tab can be left at default.

Click on **Next** on the following screens and then **Finish**.

General	
Hold Support	<input type="radio"/> None <input checked="" type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Next

7.2.2. Server Internetworking – BT NOAS side

Server Internetworking allows you to configure and manage various SIP call server-specific capabilities such as call hold and T.38. From the left hand menu select **Global Profiles** → **Server Interworking** and click on **Add Profile**.

- Enter **Profile Name: NOAS** and click on **Next**.
- Check **Hold Support= RFC2543**.
- Check **T.38 Support**.
- All other options on the General Tab can be left at default.

Click on **Next** on the following screens and then **Finish**.

General	
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

Next

7.2.3. Routing – Avaya side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages. From the left hand menu select **Global Profiles** → **Routing** and click on **Add Profile**.

- Enter **Profile Name: ToRomASM**
- Hit **Next** (not shown)
- **Next Hop Server 1: 192.168.131.186** (Session Manager IP address)
- **Next Hop Server 2: 192.168.51.46** (Session Manager backup IP address)
- Select **Routing Priority Based on Next Hop Server**
- Select **Use Next Hop for In-Dialog Messages**
- **Outgoing Transport: TCP**

Click **Finish** (not shown).

The screenshot shows the 'Add Profile' window for a Routing Profile named 'ToRomASM'. The left sidebar lists 'Routing Profiles' with 'ToRomASM' selected. The main area has a yellow header 'Click here to add a description.' and a 'Routing Profile' tab. Below is a table for 'Add Routing Rule' with the following data:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	192.168.131.186	192.168.51.46	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	TCP

7.2.4. Routing – BT NOAS side

The Routing Profile allows you to manage parameters related to routing SIP signaling messages. A routing profile must be set for Fixed and Mobile calls. From the left hand menu select **Global Profiles** → **Routing** and click on **Add Profile**.

- Enter **Profile Name: ToNOAS**
- Hit **Next**
- **Next Hop Server 1: 193.113.149.58** (IP Address provided by BT)
- **Next Hop Server 1: 193.113.149.62** (IP Address provided by BT)
- Select **Routing Priority Based on Next Hop Server**
- Select **Use Next Hop for In-Dialog Messages**
- **Outgoing Transport: UDP**
- Click **Finish** (not shown)

The screenshot shows the 'Add Profile' window for a Routing Profile named 'ToNOAS'. The left sidebar lists 'Routing Profiles' with 'ToNOAS' selected. The main area has a yellow header 'Click here to add a description.' and a 'Routing Profile' tab. Below is a table for 'Add Routing Rule' with the following data:

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	Next Hop Priority	NAPTR	SRV	Next Hop in Dialog	Ignore Route Header	Outgoing Transport
1	*	193.113.149.58	193.113.149.62	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	UDP

7.2.5. Server Configuration – Avaya Aura® Session Manager

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options. From the left hand menu select **Global Profiles** → **Server Configuration** and click on **Add Profile**.

- Enter **Profile Name**: ASM_CallServer
- On the **Add Server Configuration Profile** Tab:
- Select **Server Type**: Call Server
- **IP Address**: 192.168.131.186,192.168.51.46 (Session Manager IP Addresses)
- **Supported Transports**: Check **TCP**
- **TCP Port**:5060
- Click on **Next** for the **Authentication** and **Heartbeat** tabs.
- On the **Advanced** Tab
- Select **ToASM** for Interworking Profile
- Hit **Next**
- Click **Finish**

Edit Server Configuration Profile - General	
Server Type	Call Server
IP Addresses / Supported FQDNs Comma seperated list	192.168.131.186,192.168.51.46
Supported Transports	<input checked="" type="checkbox"/> TCP <input type="checkbox"/> UDP <input type="checkbox"/> TLS
TCP Port	5060
UDP Port	
TLS Port	
Finish	

Edit Server Configuration Profile - Advanced	
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	ToASM
Signaling Manipulation Script	None
TCP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING

Finish

7.2.6. Server Configuration– BT NOAS side

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow you to configure and manage various SIP call server-specific parameters such as TCP and UDP port assignments, server type, heartbeat signaling parameters and some advanced options. From the left-hand menu select **Global Profiles** → **Server Configuration** and click on **Add Profile**.

- **Name: ToNOAS**
- On the **Add Server Configuration Profile** Tab:
- Click on **Edit**
- Select Server Type: **Trunk Server**
- **IP Address: 193.113.149.58,193.113.149.62 (BT Trunk Server)**
- **Supported Transports: Check UDP**
- **UDP Port: 5060**
- Hit **Next**
- Click on **Next** for the **Authentication** and **Heartbeat** tabs.
- On the **Advanced** Tab
- Select **NOAS** for Interworking Profile
- Hit **Next**
- Click **Finish**

Edit Server Configuration Profile - General

Server Type	Trunk Server
IP Addresses / Supported FQDNs Comma seperated list	193.113.149.58,193.113.149.62
Supported Transports	<input type="checkbox"/> TCP <input checked="" type="checkbox"/> UDP <input type="checkbox"/> TLS
TCP Port	
UDP Port	5060
TLS Port	

Finish

Edit Server Configuration Profile - Advanced

Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input type="checkbox"/>
Interworking Profile	NOAS
Signaling Manipulation Script	None
UDP Connection Type	<input checked="" type="radio"/> SUBID <input type="radio"/> PORTID <input type="radio"/> MAPPING

Finish

7.2.7. Topology Hiding – Avaya side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. From the left-hand menu select **Global Profiles → Topology Hiding**.

- Click **default** profile and select **Clone Profile**.
- Enter Profile Name: **ASM**
- For the **To** and **Request Line** headers, select **IP/Domain** under **Criteria** and **Next Hop** under **Replace Action**.
- Click **Finish**

The screen below is a result of the details configured above.

The screenshot shows the Avaya Topology Hiding configuration interface. On the left, a sidebar lists 'Topology Hiding Profiles' with 'ToRomASM' selected. The main area has a yellow header with 'Click here to add a description.' Below it, a table titled 'Topology Hiding' is displayed. The table has four columns: 'Header', 'Criteria', 'Replace Action', and 'Overwrite Value'. Two rows are visible: 'To' and 'Request-Line', both with 'IP/Domain' in the 'Criteria' column and 'Next Hop' in the 'Replace Action' column. An 'Edit' button is located below the table.

Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Next Hop	---
Request-Line	IP/Domain	Next Hop	---

7.2.8. Topology Hiding – BT side

The **Topology Hiding** screen allows you to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks. From the left-hand menu select **Global Profiles → Topology Hiding**.

- Click **default** profile and select **Clone Profile**.
- Enter **Profile Name: ToNOAS**
- For the **To** and **Request Line** headers, select **IP/Domain** under **Criteria** and **NextHop** under **Replace Action**.
- Click **Finish**

The screen below is a result of the details configured above.

The screenshot shows the Avaya Topology Hiding configuration interface for profile 'ToNOAS'. The sidebar on the left has 'ToNOAS' selected. The main area features a yellow header with 'Click here to add a description.' Below it, a table titled 'Topology Hiding' is shown. The table has four columns: 'Header', 'Criteria', 'Replace Action', and 'Overwrite Value'. Two rows are visible: 'To' and 'Request-Line', both with 'IP/Domain' in the 'Criteria' column and 'Next Hop' in the 'Replace Action' column. An 'Edit' button is located below the table.

Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Next Hop	---
Request-Line	IP/Domain	Next Hop	---

7.3. Device Specific Settings

The **Network Management** feature allows the public and private interface addresses and state to be set. From the left-hand menu select **Device Specific Settings** → **Network Management**.

- Enter in the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces.
- Select the physical interface used in the **Interface** column.

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from **System Management**.

A1 Netmask: 255.255.255.0 A2 Netmask: [] B1 Netmask: 255.255.255.0 B2 Netmask: []

Add IP **Save Changes** **Clear Changes**

IP Address	Public IP	Gateway	Interface
192.168.130.121	[]	192.168.130.1	A1
192.168.131.133	[]	192.168.131.1	B1

Select the **Interface Configuration** Tab and use the **Toggle State** button to enable the interfaces.

Name	Administrative Status
A1	Enabled Toggle State
A2	Disabled Toggle State
B1	Enabled Toggle State
B2	Disabled Toggle State

The **Media Interfaces** feature allows the IP Address and ports to be set for transporting Media over the SIP trunk. From the left-hand menu select **Device Specific Settings → Media Interface**.

- Select **Add Media Interface**
- **Name: MediaROMASM**
- **Media IP: 192.168.131.133** (Internal Address for calls toward Session Manager)
- **Port Range: 35000-40000**
- Click **Finish**
- Select **Add Media Interface**
- **Name: MediaNOAS**
- **Media IP: 192.168.130.96** (External Address for calls toward BT trunk)
- **Port Range: 35000-40000**
- Click **Finish**
- Select **Add Media Interface**

The screen below is a result of the details configured above.

The screenshot displays the 'Media Interface' configuration page. On the left, a sidebar shows 'UC-Sec Devices' with 'RomSipera1' and 'RomSiperaNOAS' listed. The main content area has a warning banner: 'Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.' Below this is a table of configured interfaces and an 'Add Media Interface' button.

Name	Media IP	Port Range	
MediaROMASM	192.168.131.133	35000 - 40000	
MediaNOAS	192.168.130.96	35000 - 40000	

The **Signalling Interfaces** feature allows the IP Address and ports to be set for transporting Media over the SIP trunk. From the left-hand menu select **Device Specific Settings** → **Signalling Interface**.

- Select **Add Signaling Interface**
- **Name: SigROMASM**
- **Signaling IP: 192.168.131.133** (Internal Address for calls toward Session Manager)
- **TCP Port: 5060**
- **UDP Port: 5060**
- Click **Finish**
- Select **Add Media Interface**
- **Name: SigNOAS**
- **Signaling IP: 192.168.130.96** (External Address for calls toward BT)
- **TCP Port: 5060**
- **UDP Port: 5060**
- Click **Finish**

The screen below is a result of the details configured above.

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile	
SigROMASM	192.168.131.133	5060	5060	---	None	
SigNOAS	192.168.130.96	5060	5060	---	None	

The **End Point Flows** allow the Interfaces, Policies and Profiles administered to be used to transport the SIP traffic. From the left-hand menu **select Device Specific Settings** → **Endpoint Flows** and select the **Server Flows** tab. To add the settings for fixed call flow to Session Manager, click on select **Add Flow**.

- **Name: Callserver**
- **Server Configuration: ROMASM**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface: SigNOAS**
- **Signaling Interface: SigROMASM**
- **Media Interface: MediaROMASM**
- **End Point Policy Group: default-low**
- **Routing Profile: ToNOAS**
- **Topology Hiding Profile: ToROMASM**
- **File Transfer Profile: None**
- Click **Finish**

To add the settings for fixed call flow to BT, select **Add Flow**.

- **Name: TrunkServer**

- **Server Configuration: NOAS**
- **URI Group: ***
- **Transport: ***
- **Remote Subnet: ***
- **Received Interface: SigROMASM**
- **Signaling Interface: SigNOAS**
- **Media Interface: MediaNOAS**
- **End Point Policy Group: default-low**
- **Routing Profile: ToRomASM**
- **Topology Hiding Profile: ToNOAS**
- **File Transfer Profile: None**
- **Click Finish**

The screen below is a result of the details configured above.

RomSiperaNOAS													
Server Configuration: NOAS													
Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	NOAS	*	*	*	SigROMASM	SigNOAS	MediaNOAS	default-low	ToRomASM	ToNOAS	None		
Server Configuration: ROMASM													
Priority	Flow Name	URI Group	Transport	Remote Subnet	Received Interface	Signaling Interface	Media Interface	End Point Policy Group	Routing Profile	Topology Hiding Profile	File Transfer Profile		
1	to_and_from_ASM	*	*	*	SigNOAS	SigROMASM	MediaROMASM	default-low	ToNOAS	ToRomASM	None		

8. Service Provider Configuration

The configuration of the BT equipment used to support the BT SIP trunk service is outside of the scope for these Application Notes and will not be covered. To obtain further information on BT equipment and system configuration please contact an authorized BT representative using the contact details provided in **Section 2.3**.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager Home Tab click on Session Manager and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.

This is the SIP Entity link to the Communication Server 1000E:

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager x Routing x Home

Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring

SIP Entity, Entity Link Connection Status Help ?

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

The following errors have occurred:
Unable to access SIP monitoring data from Session Manager, Romford SM 6.1 - cannot connect to server.

All Entity Links to SIP Entity: [PRIM_SS_LEADER](#)

Summary View

2 Items Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	Leeds SM6.1	192.168.51.34	5061	TLS	Up	200 OK	Up
Show	Leeds SM6.1	192.168.51.34	5060	TCP	Up	200 OK	Up

This is the SIP Entity link to the Avaya Session Border Controller for Enterprise:

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager x Routing x Home

Home / Elements / Session Manager / System Status / SIP Entity Monitoring - SIP Entity Monitoring

SIP Entity, Entity Link Connection Status Help ?

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

The following errors have occurred:
Unable to access SIP monitoring data from Session Manager, Romford SM 6.1 - cannot connect to server.

All Entity Links to SIP Entity: [2nd_Romford_AASBC6.0](#)

Summary View

2 Items Refresh Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	Leeds SM6.1	192.168.131.133	5060	UDP	Up	200 OK	Up
Show	Leeds SM6.1	192.168.131.133	5060	TCP	Up	200 OK	Up

2. From the Communication Server 1000E system terminal; load overlay 32 and run the command 'stat vtrm <cust> <x>' where 'cust' is the customer number (usually 0) and 'x' is a previously configured SIP trunk route. Confirm all channels on the trunk group display idle registered.

```
stat vtrm 0 100

*****
STATUS OF VTRL IP TRUNK ROUTE AND MBRs
*****

=====
CUST ROUTE PROTOCOL CALL DIRCTN
0 100 SIP IN AND OUT

DCH 50 SSRC TOTAL 2048 SSRC USED 77 SSRC AVAILABLE 1971

MBR STATUS

IDLE UNREGISTERED 0
IDLE REGISTERED 15
BUSY 0
MBSY 0
DSBL UNREGISTERED 0
DSBL REGISTERED 0
LCKO 0
```

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call remains active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya Session Border Controller for Enterprise to the BT SIP Trunk Service. BT SIP Trunk Service is a SIP-based Voice over IP solution providing businesses with a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6, June 2010.
- [2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.
- [3] *Avaya Communication Server 1000E Installation and Commissioning*, November 2010, Document Number NN43041-310.
- [4] *Feature Listing Reference Avaya Communication Server 1000*, November 2010, Document Number NN43001-111, 05.01.
- [5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.
- [6] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, March 2011, Document Number 03-603324.
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

Appendix A – Communication Server 1000 Software

Communication Server 1000E call server patches and plug_ins

```

18/08/11 10:33:16
TID: 008808096

VERSION 4021

System type is - Communication Server 1000E/CP PM

CP PM - Pentium M 1.4 GHz

IPMGs Registered:          4
IPMGs Unregistered:       0
IPMGs Configured/unregistered: 2

RELEASE 7
ISSUE 50 Q +
IDLE SET DISPLAY Avaya 7.5
DepList 1: core Issue: 02 (created: 2010-11-30 15:12:45 (est))

MDP>LAST SUCCESSFUL MDP REFRESH :2010-12-06 15:33:54 (Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-12-01 08:31:36 (est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE

LOADWARE VERSION: PSWV 100
INSTALLED LOADWARE PEPS : 0
ENABLED PLUGINS : 0
    
```

Communication Server 1000E call server deplists

```

VERSION 4021
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2012-02-03 09:32:18 (est))

IN-SERVICE PEPS
PAT# CR #          PATCH REF #    NAME          DATE          FILENAME          SPECINS
000 wi00832106      ISS1:10F1    p30550_1      19/02/2012    p30550_1.cpm     NO
001 wi00856991      ISS1:10F1    p17588_1      19/02/2012    p17588_1.cpm     NO
002 wi00950857      ISS1:10F1    p24307_1      19/02/2012    p24307_1.cpm     NO
003 wi00881777      ISS1:10F1    p25747_1      19/02/2012    p25747_1.cpm     NO
004 wi00905660      ISS1:10F1    p27968_1      19/02/2012    p27968_1.cpm     NO
005 wi00688381      ISS1:10F1    p30104_1      19/02/2012    p30104_1.cpm     NO
006 WI00839794      ISS1:10F1    p28647_1      19/02/2012    p28647_1.cpm     NO
007 wi00896680      ISS1:10F1    p30357_1      19/02/2012    p30357_1.cpm     NO
008 wi00825488      ISS1:10F1    p30371_1      19/02/2012    p30371_1.cpm     NO
009 wi00825486      ISS1:10F1    p30382_1      19/02/2012    p30382_1.cpm     NO
010 wi00961267      ISS1:10F1    p30288_1      19/02/2012    p30288_1.cpm     NO
011 wi00879322      ISS1:10F1    p30954_1      19/02/2012    p30954_1.cpm     NO
012 wi00897176      ISS1:10F1    p30418_1      19/02/2012    p30418_1.cpm     NO
013 wi00903381      ISS1:10F1    p30421_1      19/02/2012    p30421_1.cpm     NO
014 wi00854130      ISS1:10F1    p30443_1      19/02/2012    p30443_1.cpm     NO
015 wi00824257      ISS1:10F1    p30447_1      19/02/2012    p30447_1.cpm     NO
016 wi00836182      ISS1:10F1    p30450_1      19/02/2012    p30450_1.cpm     NO
017 wi00826075      ISS1:10F1    p30452_1      19/02/2012    p30452_1.cpm     NO
018 WI00900668      ISS1:10F1    p30456_1      19/02/2012    p30456_1.cpm     NO
019 wi00854150      ISS1:10F1    p30468_1      19/02/2012    p30468_1.cpm     NO
020 wi00827950      ISS2:10F1    p30471_2      19/02/2012    p30471_2.cpm     NO
021 WI00836334      ISS1:10F1    p30481_1      19/02/2012    p30481_1.cpm     NO
022 wi00833910      ISS2:10F1    p30492_2      19/02/2012    p30492_2.cpm     NO
023 wi00853031      ISS1:10F1    p30531_1      19/02/2012    p30531_1.cpm     NO
024 wi00877367      ISS1:10F1    p30534_1      19/02/2012    p30534_1.cpm     NO
    
```

025	wi00834382	ISS1:10F1	p30548_1	19/02/2012	p30548_1.cpm	NO
026	WI00836292	ISS1:10F1	p30554_1	19/02/2012	p30554_1.cpm	NO
027	wi00835294	ISS1:10F1	p30565_1	19/02/2012	p30565_1.cpm	NO
028	wi00856702	ISS1:10F1	p30573_1	19/02/2012	p30573_1.cpm	NO
029	wi00838073	ISS1:10F1	p30588_1	19/02/2012	p30588_1.cpm	NO
030	wi00839255	ISS1:10F1	p30591_1	19/02/2012	p30591_1.cpm	NO
031	wi00854415	ISS1:10F1	p30593_1	19/02/2012	p30593_1.cpm	NO
032	wi00836981	ISS1:10F1	p30613_1	19/02/2012	p30613_1.cpm	NO
033	wi00841980	ISS1:10F1	p30618_1	19/02/2012	p30618_1.cpm	NO
034	wi00839821	ISS1:10F1	p30619_1	19/02/2012	p30619_1.cpm	NO
035	wi00842409	ISS1:10F1	p30621_1	19/02/2012	p30621_1.cpm	NO
036	WI00853473	ISS1:10F1	p30625_1	19/02/2012	p30625_1.cpm	NO
037	WI00843571	ISS1:10F1	p30627_1	19/02/2012	p30627_1.cpm	NO
038	wi00867905	ISS1:10F1	p30640_1	19/02/2012	p30640_1.cpm	NO
039	wi00852389	ISS1:10F1	p30641_1	19/02/2012	p30641_1.cpm	NO
040	wi00859123	ISS1:10F1	p30648_1	19/02/2012	p30648_1.cpm	NO
041	wi00869695	ISS1:10F1	p30654_1	19/02/2012	p30654_1.cpm	NO
042	WI00900213	ISS1:10F1	p30656_1	19/02/2012	p30656_1.cpm	NO
043	wi00897096	ISS1:10F1	p30676_1	19/02/2012	p30676_1.cpm	NO
044	wi00859449	ISS1:10F1	p30694_1	19/02/2012	p30694_1.cpm	NO
045	wi00839134	ISS1:10F1	p30698_1	19/02/2012	p30698_1.cpm	YES
046	wi00953811	ISS1:10F1	p31002_1	19/02/2012	p31002_1.cpm	NO
047	wi00852365	ISS1:10F1	p30707_1	19/02/2012	p30707_1.cpm	NO
048	wi00850521	ISS1:10F1	p30709_1	19/02/2012	p30709_1.cpm	YES
049	wi00841273	ISS1:10F1	p30713_1	19/02/2012	p30713_1.cpm	NO
050	wi00853178	ISS1:10F1	p30719_1	19/02/2012	p30719_1.cpm	NO
051	wi00843623	ISS1:10F1	p30731_1	19/02/2012	p30731_1.cpm	YES
052	wi00856410	ISS1:10F1	p30749_1	19/02/2012	p30749_1.cpm	NO
053	WI00889786	ISS1:10F1	p30750_1	19/02/2012	p30750_1.cpm	NO
054	wi00857566	ISS1:10F1	p30766_1	19/02/2012	p30766_1.cpm	NO
055	wi00840590	ISS1:10F1	p30767_1	19/02/2012	p30767_1.cpm	NO
056	wi00871969	ISS1:10F1	p30768_1	19/02/2012	p30768_1.cpm	NO
057	wi00857362	ISS1:10F1	p30782_1	19/02/2012	p30782_1.cpm	NO
058	wi00863876	ISS1:10F1	p30787_1	19/02/2012	p30787_1.cpm	NO
059	wi00860279	ISS1:10F1	p30789_1	19/02/2012	p30789_1.cpm	NO
060	wi00859305	ISS1:10F1	p30792_1	19/02/2012	p30792_1.cpm	NO
061	wi00925141	ISS1:10F1	p30802_1	19/02/2012	p30802_1.cpm	NO
062	wi00896394	ISS1:10F1	p30807_1	19/02/2012	p30807_1.cpm	NO
063	wi00899584	ISS1:10F1	p30809_1	19/02/2012	p30809_1.cpm	NO
064	wi00858335	ISS1:10F1	p30819_1	19/02/2012	p30819_1.cpm	NO
065	wi00873382	ISS1:10F1	p30832_1	19/02/2012	p30832_1.cpm	NO
066	wi00932942	ISS1:10F1	p30843_1	19/02/2012	p30843_1.cpm	NO
067	wi00877442	ISS1:10F1	p30844_1	19/02/2012	p30844_1.cpm	NO
068	wi00869243	ISS1:10F1	p30848_1	19/02/2012	p30848_1.cpm	NO
069	wi00871739	ISS1:10F1	p30856_1	19/02/2012	p30856_1.cpm	NO
070	wi00890475	p30952	p31048_1	19/02/2012	p31048_1.cpm	NO
071	wi00896420	ISS1:10F1	p30867_1	19/02/2012	p30867_1.cpm	NO
072	wi00862574	iss1:lof1	p30870_1	19/02/2012	p30870_1.cpm	NO
073	wi00877592	ISS1:10F1	p30880_1	19/02/2012	p30880_1.cpm	NO
074	wi00938555	ISS1:10F1	p30881_1	19/02/2012	p30881_1.cpm	YES
075	wi00865477	ISS1:10F1	p30890_1	19/02/2012	p30890_1.cpm	YES
076	wi00865477	ISS1:10F1	p30891_1	19/02/2012	p30891_1.cpm	YES
077	wi00865477	ISS1:10F1	p30892_1	19/02/2012	p30892_1.cpm	YES
078	wi00865477	ISS1:10F1	p30893_1	19/02/2012	p30893_1.cpm	YES
079	wi00865477	ISS1:10F1	p30894_1	19/02/2012	p30894_1.cpm	YES
080	wi00865477	ISS1:10F1	p30895_1	19/02/2012	p30895_1.cpm	YES
081	wi00865477	ISS1:10F1	p30896_1	19/02/2012	p30896_1.cpm	YES
082	wi00865477	ISS1:10F1	p30897_1	19/02/2012	p30897_1.cpm	YES
083	wi00865477	ISS1:10F1	p30898_1	19/02/2012	p30898_1.cpm	YES
084	wi00875701	ISS1:10F1	p30942_1	19/02/2012	p30942_1.cpm	NO
085	wi00875425	ISS1:10F1	p30943_1	19/02/2012	p30943_1.cpm	NO
086	wi00903369	ISS1:10F1	p31165_1	19/02/2012	p31165_1.cpm	NO
087	wi00946282	ISS1:10F1	p31204_1	19/02/2012	p31204_1.cpm	NO
088	wi00883604	ISS1:10F1	p30973_1	19/02/2012	p30973_1.cpm	NO
089	wi00882884	ISS1:10F1	p30975_1	19/02/2012	p30975_1.cpm	NO
090	wi00880836	ISS1:10F1	p30976_1	19/02/2012	p30976_1.cpm	NO
091	wi00880386	ISS1:10F1	p30977_1	19/02/2012	p30977_1.cpm	NO
092	wi00925208	ISS1:10F1	p30986_1	19/02/2012	p30986_1.cpm	NO
093	WI00927300	ISS1:10F1	p30999_1	19/02/2012	p30999_1.cpm	NO
094	wi00884699	ISS1:10F1	p31000_1	19/02/2012	p31000_1.cpm	YES

095	wi00900096	ISS1:10F1	p31006_1	19/02/2012	p31006_1.cpm	NO
096	wi00879526	ISS1:10F1	p31007_1	19/02/2012	p31007_1.cpm	NO
097	wi00886321	ISS1:10F1	p31009_1	19/02/2012	p31009_1.cpm	NO
098	wi00882293	ISS1:10F1	p31010_1	19/02/2012	p31010_1.cpm	NO
099	wi00887744	ISS2:10F1	p31026_2	19/02/2012	p31026_2.cpm	NO
100	WI00928455	ISS1:10F1	p31297_1	19/02/2012	p31297_1.cpm	NO
101	wi00889088	ISS1:10F1	p31036_1	19/02/2012	p31036_1.cpm	NO
102	wi00890036	ISS1:10F1	p31044_1	19/02/2012	p31044_1.cpm	NO
103	wi00891626	ISS1:10F1	p31051_1	19/02/2012	p31051_1.cpm	YES
104	wi00880221	ISS1:10F1	p31054_1	19/02/2012	p31054_1.cpm	NO
105	wi00877365	ISS1:10F1	p31060_1	19/02/2012	p31060_1.cpm	NO
106	wi00932948	ISS1:10F1	p31077_1	19/02/2012	p31077_1.cpm	NO
107	wi00894243	ISS1:10F1	p31087_1	19/02/2012	p31087_1.cpm	NO
108	wi00893131	ISS1:10F1	p31089_1	19/02/2012	p31089_1.cpm	NO
109	wi00894443	ISS1:10F1	p31093_1	19/02/2012	p31093_1.cpm	NO
110	wi00895090	ISS1:10F1	p31105_1	19/02/2012	p31105_1.cpm	NO
111	wi00895181	ISS1:10F1	p31106_1	19/02/2012	p31106_1.cpm	NO
112	wi00932958	ISS1:10F1	p31115_1	19/02/2012	p31115_1.cpm	NO
113	wi00897082	ISS1:10F1	p31124_1	19/02/2012	p31124_1.cpm	NO
114	wi00937114	ISS1:10F1	p31310_1	19/02/2012	p31310_1.cpm	NO
115	wi00898327	ISS1:10F1	p31136_1	19/02/2012	p31136_1.cpm	NO
116	wi00900766	ISS1:10F1	p31159_1	19/02/2012	p31159_1.cpm	NO
117	wi00868729	ISS1:10F1	p31163_1	19/02/2012	p31163_1.cpm	NO
118	wi00903085	ISS1:10F1	p31164_1	19/02/2012	p31164_1.cpm	NO
119	wi00903437	ISS1:10F1	p31167_1	19/02/2012	p31167_1.cpm	NO
120	wi00855423	ISS1:10F1	p31328_1	19/02/2012	p31328_1.cpm	YES
121	wi00905297	ISS1:10F1	p31195_1	19/02/2012	p31195_1.cpm	NO
122	wi00905600	ISS1:10F1	p31201_1	19/02/2012	p31201_1.cpm	NO
123	wi00906022	ISS1:10F1	p31202_1	19/02/2012	p31202_1.cpm	NO
124	wi00906098	ISS1:10F1	p31203_1	19/02/2012	p31203_1.cpm	YES
125	wi00906163	ISS1:10F1	p31205_1	19/02/2012	p31205_1.cpm	NO
126	wi00946558	ISS1:10F1	p31358_1	19/02/2012	p31358_1.cpm	NO
127	wi00907403	ISS1:10F1	p31225_1	19/02/2012	p31225_1.cpm	NO
128	wi00948274	ISS1:10F1	p31365_1	19/02/2012	p31365_1.cpm	NO
129	wi00907697	ISS1:10F1	p31227_1	19/02/2012	p31227_1.cpm	NO
130	wi00907707	ISS1:10F1	p31228_1	19/02/2012	p31228_1.cpm	NO
131	wi00892954	ISS1:10F1	p31378_1	19/02/2012	p31378_1.cpm	NO
132	wi00908598	ISS1:10F1	p31235_1	19/02/2012	p31235_1.cpm	NO
133	wi00908933	ISS1:10F1	p31239_1	19/02/2012	p31239_1.cpm	NO
134	wi00936714	ISS1:10F1	p31379_1	19/02/2012	p31379_1.cpm	NO
135	wi00921295	ISS1:10F1	p31265_1	19/02/2012	p31265_1.cpm	NO
136	wi00921340	ISS1:10F1	p31266_1	19/02/2012	p31266_1.cpm	NO
137	wi00923899	ISS1:10F1	p31270_1	19/02/2012	p31270_1.cpm	NO
138	wi00898200	ISS1:10F1	p31274_1	19/02/2012	p31274_1.cpm	NO
139	wi00929140	ISS1:10F1	p31284_1	19/02/2012	p31284_1.cpm	NO
140	wi00927321	ISS1:10F1	p31286_1	19/02/2012	p31286_1.cpm	YES
141	wi00932204	ISS2:10F1	p31305_2	19/02/2012	p31305_2.cpm	NO
142	wi00925033	ISS1:10F1	p31320_1	19/02/2012	p31320_1.cpm	NO
143	wi00930864	ISS1:10F1	p31325_1	19/02/2012	p31325_1.cpm	NO
144	wi00909476	ISS1:10F1	p31340_1	19/02/2012	p31340_1.cpm	NO
145	wi00931028	ISS1:10F1	p31354_1	19/02/2012	p31354_1.cpm	YES
146	wi00941500	ISS1:10F1	p31394_1	19/02/2012	p31394_1.cpm	NO
147	wi00932929	ISS1:10F1	p31392_1	19/02/2012	p31392_1.cpm	YES
148	wi00927678	ISS1:10F1	p31399_1	19/02/2012	p31399_1.cpm	NO
149	wi00943172	ISS1:10F1	p31402_1	19/02/2012	p31402_1.cpm	NO
150	wi00948931	ISS1:10F1	p31407_1	19/02/2012	p31407_1.cpm	NO
151	wi00942734	ISS1:10F1	p31409_1	19/02/2012	p31409_1.cpm	NO
152	wi00949273	ISS1:10F1	p31411_1	19/02/2012	p31411_1.cpm	NO
153	wi00945533	ISS1:10F1	p31421_1	19/02/2012	p31421_1.cpm	YES
154	wi00946477	ISS1:10F1	p31426_1	19/02/2012	p31426_1.cpm	NO
155	wi00946681	ISS1:10F1	p31428_1	19/02/2012	p31428_1.cpm	NO
156	wi00946876	ISS1:10F1	p31430_1	19/02/2012	p31430_1.cpm	NO
157	wi00949627	ISS1:10F1	p31462_1	19/02/2012	p31462_1.cpm	NO
158	wi00951427	ISS1:10F1	p31478_1	19/02/2012	p31478_1.cpm	NO
159	wi00951837	ISS1:10F1	p31485_1	19/02/2012	p31485_1.cpm	NO
160	wi00951925	ISS1:10F1	p31486_1	19/02/2012	p31486_1.cpm	NO
161	wi00956885	ISS1:10F1	p31489_1	19/02/2012	p31489_1.cpm	NO
162	wi00953900	ISS1:10F1	p31494_1	19/02/2012	p31494_1.cpm	NO
163	wi00953902	ISS1:10F1	p31499_1	19/02/2012	p31499_1.cpm	NO
164	wi00955541	ISS1:10F1	p31501_1	19/02/2012	p31501_1.cpm	NO

165	wi00925989	ISS1:10F1	p31514_1	19/02/2012	p31514_1.cpm	NO
166	wi00943748	ISS1:10F1	p31516_1	19/02/2012	p31516_1.cpm	NO
167	wi00957252	ISS1:10F1	p31530_1	19/02/2012	p31530_1.cpm	NO
168	wi00958682	ISS1:10F1	p31540_1	19/02/2012	p31540_1.cpm	NO
169	wi00958776	ISS1:10F1	p31542_1	19/02/2012	p31542_1.cpm	YES
170	wi00959854	ISS1:10F1	p31556_1	19/02/2012	p31556_1.cpm	NO
171	wi00960133	ISS2:10F1	p31557_2	19/02/2012	p31557_2.cpm	NO
172	wi00959820	ISS1:10F1	p31562_1	19/02/2012	p31562_1.cpm	NO
173	wi00957141	ISS1:10F1	p31579_1	19/02/2012	p31579_1.cpm	NO
174	wi00962211	ISS1:10F1	p31580_1	19/02/2012	p31580_1.cpm	NO
175	wi00962557	ISS1:10F1	p31581_1	19/02/2012	p31581_1.cpm	NO
176	wi00962955	ISS1:10F1	p31585_1	19/02/2012	p31585_1.cpm	NO
177	wi00965724	ISS1:10F1	p31606_1	19/02/2012	p31606_1.cpm	NO
MDP>LAST SUCCESSFUL MDP REFRESH :2012-02-19 09:06:34 (Local Time)						
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2012-02-03 09:32:18 (est)						

Communication Server 1000E signaling server service updates

Product Release: 7.50.17.00

In system patches: 1

PATCH#	NAME	IN_SERVICE	DATE	SPECINS	TYPE	RPM
2	p30526 1	Yes	19/02/12	NO	FRU	cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 22

PATCH#	IN SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	12/12/11	NO	YES	cs1000-baseWeb-7.50.17.16-1.i386.001
1	Yes	18/02/12	NO	YES	cs1000-csmWeb-7.50.17.16-3.i386.000
3	Yes	17/02/12	NO	YES	cs1000-linuxbase-7.50.17.16-6.i386.000
4	Yes	18/02/12	NO	YES	cs1000-mscAnnc-7.50.17.16-1.i386.000
5	Yes	18/02/12	NO	YES	cs1000-mscTone-7.50.17.16-1.i386.000
6	Yes	18/02/12	NO	YES	cs1000-mscMusc-7.50.17.16-2.i386.000
7	Yes	18/02/12	NO	YES	cs1000-dmWeb-7.50.17.16-2.i386.000
8	Yes	18/02/12	NO	YES	cs1000-sps-7.50.17.16-2.i386.000
9	Yes	14/12/11	NO	YES	cs1000-dbcom-7.50.17-02.i386.000
10	Yes	19/02/12	NO	YES	cs1000-tps-7.50.17.16-11.i386.000
11	Yes	14/12/11	NO	YES	cs1000-shared-pbx-7.50.17.16-1.i386.000
12	Yes	14/12/11	NO	YES	cs1000-kcv-7.50.17.16-1.i386.000
13	Yes	19/02/12	NO	yes	avaya-cs1000-cnd-4.0.20-00.i386.000
14	Yes	14/12/11	NO	YES	cs1000-ipsec-7.50.17.16-1.i386.000
15	Yes	19/02/12	NO	YES	cs1000-emWeb 6-0-7.50.17.16-16.i386.000
16	Yes	19/02/12	NO	YES	cs1000-Jboss-Quantum-7.50.17.16-10.i386.000
17	Yes	17/03/12	NO	YES	cs1000-vtrk-7.50.17.16-43.i386.000
18	Yes	14/12/11	NO	YES	ipsec-tools-0.6.5-14.el5.3 avaya 1.i386.000
19	Yes	14/12/11	NO	YES	spiritAgent-6.1-1.0.0.108.208.i386.000
20	Yes	14/12/11	NO	YES	cs1000-EmCentralLogic-7.50.17.16-1.i386.000
25	Yes	18/02/12	NO	YES	cs1000-ftrpkg-7.50.17.16-7.i386.000
26	Yes	18/02/12	NO	YES	cs1000-bcc-7.50.17.16-46.i386.000

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